(12) INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(19) World Intellectual Property Organization
International Bureau

(43) International Publication Date
27 September 2001 (27.09.2001)

PCT

(10) International Publication Number
WO 01/72022 A1

(51) International Patent Classification[7]: H04M 3/42

(21) International Application Number: PCT/US01/08082

(22) International Filing Date: 14 March 2001 (14.03.2001)

(25) Filing Language: English

(26) Publication Language: English

(30) Priority Data:
09/528,549 20 March 2000 (20.03.2000) US

(71) Applicant: EYAK, INC. [—/US]; 70 Franklin Street, Boston, MA 02110 (US).

(72) Inventors: FLOWERS, Jeffrey, C.; 391 Ocean Avenue, Marblehead, MA 01945 (US). KOSCHELLA, James, J.; 70 Boston Avenue, Somerville, MA 02144 (US).

(74) Agent: SCHAEFER, Ira, J.; Clifford Chance Rogers & Wells LLP, 200 Park Avenue, New York, NY 10166-0153 (US).

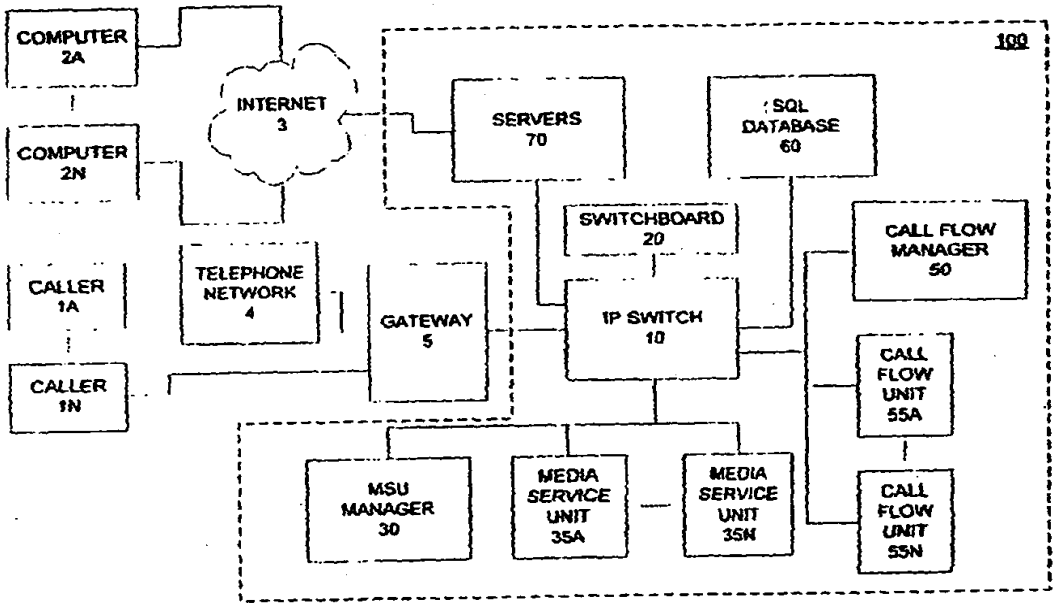
(81) Designated States *(national)*: AE, AG, AL, AM, AT, AU, AZ, BA, BB, BG, BR, BY, BZ, CA, CH, CN, CR, CU, CZ, DE, DK, DM, DZ, EE, ES, FI, GB, GD, GE, GH, GM, HR, HU, ID, IL, IN, IS, JP, KE, KG, KP, KR, KZ, LC, LK, LR, LS, LT, LU, LV, MA, MD, MG, MK, MN, MW, MX, MZ, NO, NZ, PL, PT, RO, RU, SD, SE, SG, SI, SK, SL, TJ, TM, TR, TT, TZ, UA, UG, UZ, VN, YU, ZA, ZW.

(84) Designated States *(regional)*: ARIPO patent (GH, GM, KE, LS, MW, MZ, SD, SL, SZ, TZ, UG, ZW), Eurasian patent (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European patent (AT, BE, CH, CY, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE, TR), OAPI patent (BF, BJ, CF, CG, CI, CM, GA, GN, GW, ML, MR, NE, SN, TD, TG).

Published:
— *with international search report*

For two-letter codes and other abbreviations, refer to the "Guidance Notes on Codes and Abbreviations" appearing at the beginning of each regular issue of the PCT Gazette.

(54) Title: METHOD AND SYSTEM FOR PROVIDING AUDIO CONFERENCING SERVICES



(57) Abstract: A method and system (100) for providing audio conferencing services, comprises receiving Internet protocol packets containing call set-up information and Internet protocol packets containing call content information for at least one first participant at a first telephone number and setting up a conference call including the at least one first participant (1A...1N, 2A...2N) and at least one additional participant (1A...1N, 2A...2N) at a second telephone number based solely on the Internet protocol packets.

WO 01/72022 A1

METHOD AND SYSTEM FOR PROVIDING AUDIO CONFERENCING SERVICES

BACKGROUND OF THE INVENTION

The present invention relates to a method and system for providing audio conferencing services and to an improved teleconferencing bridge.

Audio conferencing services allow groups of people to communicate via telephone. Any person with access to a public telephone network can place a call to or receive a call from a centralized audio conferencing network that allows two or more people to talk as if they were in a conference. This is accomplished through the use of conventional teleconferencing switches such as an AT&T 5ESS switch which allows two or more callers to communicate with each other.

Conventional teleconferencing bridges use analog audio conferencing switches that receive an incoming audio signal from one or more callers participating in a conference and send that signal or combined signals of multiple callers to the participants of the telephone conference. One major disadvantage of conventional teleconferencing bridges and the switches that are used

therein is that they are expensive to build and maintain and are not scalable to add additional capacity.

Another disadvantage of conventional teleconferencing bridges and switches is that they are not suitable to take advantage of the convergence of telephone networks and the Internet.

It is therefore desirable to have an audio conferencing bridge that is inexpensive to build and which is easily scalable both from a hardware point of view and a cost point of view compared to conventional analog bridges.

Another object of the present invention is to provide an improved audio conferencing method.

A further object of the present invention is to provide an improved audio conferencing system.

SUMMARY OF THE INVENTION

These and other objects and advantages of the present invention are achieved in accordance with the present invention by a method and system for providing audio conferencing services and to a teleconferencing bridge in accordance with the present invention.

In accordance with the present invention, a person with access to a telephone network, either landline or wireless or persons having direct connection to the system according to the present invention, can place a call to or receive a call from the telephone conferencing bridge that allows two or more people to talk as if they were in a conference.

The teleconferencing bridge in accordance with present invention is connected to a voice over Internet Protocol (VOIP) gateway which is connected to one or more telephone networks including a local telephone network of a local exchange carrier and which can be directly accessed by certain users. The gateway converts signals to Internet Protocol (IP) packets comprising either call signaling information or call content information. The call signaling information provides the call set up signals that notify the bridge that a call is coming in and allows the bridge to notify a local exchange carrier that the bridge would like to place an outgoing call.

The teleconferencing bridge includes an IP switch which receives the call signaling IP packets and the call content IP packets and which directs packets to other components of the bridge. The bridge includes a switchboard that allows the IP switch to receive and send calls through the gateway to the telephone networks. The bridge also includes a call flow manager which controls call flow units. The call flow manager receives the call signaling information and runs scripts in Visual Basic or other languages to lead a user through a call to perform various functions of the method and system for providing audio conferencing services. The call flow units are multiple processors which carry out the scripts for each of the individual conferences that are being set-up by the system. The bridge further comprises a media service unit manager, and a plurality of media service units. The media service unit manager controls the media service units to handle the various lines to set-up the conference calls and then receives the call content IP packets to direct them via the IP switch to the correct destinations. The bridge further includes an SQL database which contains the

necessary information regarding users of the method and system such as membership identification, passwords, billing plans and the like.

The bridge also preferably includes servers which connect the IP switch to the Internet so that visual information, in additional to audio information can be shared between the computers of various callers in a conference.

In one embodiment of the invention, the method of providing audio conferencing services comprising the steps of receiving Internet protocol packets containing call set-up information and Internet protocol packets containing call content information for at least one first participant at a first telephone number and setting up a conference call including at least one first participant and at least one additional participant at a second telephone number based solely on the Internet protocol packets.

The step of setting up the conference call can comprise channeling all of the Internet protocol packets through an Internet protocol switch. Additionally the step

of setting up the conference call can further comprise receiving the call set-up information Internet protocol packets in a call flow manager to lead the first participant through a set-up procedure and receiving the content information Internet protocol packets in a media service unit manager to provide a flow of content information between the first participant and additional participants.

The step of leading the first participant can comprise running at least one of a plurality of scripts in the call flow manager and the step of providing a flow of content information can comprise controlling a plurality of media service units each handling a plurality of lines corresponding to the first and second telephone numbers.

The first participant can be a host of the conference call and the host can be provided with conference call service options for a conference call. For example, the host can be asked to enter a telephone number of an additional participant and the telephone number of the additional participant is called to add same to the conference call, the host is permitted to mute and unmute

at least one of the participants, the host is permitted to mute and unmute all of the additional participants, and the host is permitted to query as to the number of participants in the conference call and the number is returned to the host.

The method can further comprise providing a web site relating to the audio conferencing services, allowing participants to access the web site and display images thereon and providing images from a display of at least one of the participants on a display of at least one of the other participants of a conference call. The method can also provide conference call service options on a display of at least one participant and permit the at least one participant to select a conference call service option from the display. The options can comprise at least one of dialing out to additional participants, muting at least one of the participants and determining the number of participants in the conference call.

The system for providing audio conferencing services according to the invention comprises a conference bridge having an Internet protocol switch receptive of

Internet protocol packets containing call set-up information and Internet protocol packets containing call content information for at least one first participant at a first telephone number and at least one processor for setting up a conference call including the at least one first participant and at least one additional participant at a second telephone number based solely on the Internet protocol packets. The conference bridge can channel all of the Internet protocol packets through the Internet protocol switch.

The at least one processor can comprise a call flow manager for receiving the call set-up information Internet protocol packets to lead the first participant through a set-up procedure and a media service unit manager for receiving the content information Internet protocol packets to provide a flow of content information between the first participant and additional participants. The call flow manager can run at least one of a plurality of scripts to lead the first participant. The system can further comprise a plurality of media service units each handling a plurality of lines corresponding to the first

BNSDOCID: <WO 0172022A1 | >

and second telephone numbers to provide the flow of content information.

The present invention also comprises a conference bridge comprising an Internet protocol based telephone switch, a switchboard for receiving information from the switch to receive calls and dial out, a call flow manager receptive of call set up Internet protocol packets from the switch for leading a caller through a conference call set up procedure, and a media service unit manager receptive of call content Internet protocol packets from the switch for providing call content information flow between conference call participants.

The media service unit manager can manage a plurality of media service units which are scalable in number to handle an increased number of lines.

In accordance with a further aspect of the present invention, the operator of the conference bridge can bill for the audio conferencing services in a number of ways. For example, the operator of the bridge can bill the host for each participant minute. Alternatively, the operator of the bridge can bill each participant per minute

of connection to a conference. Additionally, the conference bridge can play advertisements at selective times during the conference call or show advertisements on the computer screen of the participant. Moreover, the bridge according to the present invention can have a local telephone company associated therewith. Thus each person calling the conference number will not have to pay for the conference, since the bridge will be receiving the payment from the long distance companies for terminating each of the long distance calls.

These and other features of the present invention will become apparent from the following detailed description of invention taken with the drawings, wherein:

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 is a schematic view of an audio conferencing system in accordance with the present invention for carrying out the method of the present invention;

Figures 2-10 are flow charts of methods for providing audio conferencing services in accordance with the present invention; and

Figure 11 is a screen for controlling a conference from a computer.

DETAILED DESCRIPTION OF THE INVENTION

Figure 1 illustrates the teleconferencing bridge 100 of the present invention as it is connected to carry out a system and method for providing audio conferencing services. As shown, the bridge 100 is connected to the Internet 3 and to a gateway 5 which provides IP packets of call set up information and IP packets of call content information received from callers 1A-1N over telephone network 4 or directly into the gateway as shown. The gateway is preferably a Sonus Networks GSX 9000 The bridge 100 also receives information from the Internet from computers 2A-2N which are preferably associated with various callers.

BNSDOCID: <WO___0172022A1_I_>

The bridge 100 includes as the hub component, an IP switch 10 which is preferably a Cisco Systems 6509 switch. Connected to the IP switch 10 is a switchboard 20 which enables the IP switch to receive calls via the gateway 5 from the telephone network 4 and from callers 1A-1N and to call out to callers. Also connected to the IP switch 10 is the media service unit manager 30 which controls a plurality of media service units 35A-35N. The call flow manager 50 is also connected to the IP switch 10 and controls call flow units 55A-55N. An SQL database 60 is connected to the IP switch as are servers 70 which in turn are connected to the Internet 3. The database and servers are preferably Dell 6350 or 8350 servers.

The call flow manager 50 receives the call signaling information packets from the IP switch and controls the call flow units to carry out Visual Basic (or other language) scripts which prompt a caller to lead the caller through a series of menus to enable the caller to obtain the desired function in accordance with the method of the present invention. The call flow manager and call flow units are preferably implemented by a multiprocessor

having a plurality of Pentium III processors such as a Dell 6350 or 8350 server capable of carrying out the functions with a high degree of redundancy or cluster processing so that the failure of one processor will not prevent the functions of the call flow manager and call flow units from being carried out.

The media service unit manager 30 controls media service units 35A-35N in order to process the call content IP packets so that the packets are assembled for each conference at the appropriate destination of the conference. The media service manager and media service units also handle information received from computers 2A-2N over the Internet 3 via the servers 70 so that information shown on the monitor of one computer can be conferenced and thereby shown on the monitors of other computers of callers in the same conference. The IP switch is also able to receive control information from the computers 2A-2N for controlling the call flow via the call flow manager 50, as will be discussed hereinafter.

The media service unit (MSU) manager as well as the IP switch and switchboard have the same multiprocessor or cluster processing configuration as described above. The MSU manager and switchboard are preferably Dell 6350 or 8350 servers. In a preferred embodiment, the media service units each comprise 10 central processing units which are Pentium III microprocessors such as Dell 4350 servers which are each able to handle 200 lines. Thus a single media service unit can handle 2,000 lines and these units can be connected as shown. Thus in order to scale up the system to handle more conference lines, one need only add media service units under the control of the media service unit manager.

The media service unit manager determines which media service unit has free space for controlling the content of the conference.

The switchboard 20 communicates with the gateway 5 through the IP switch 10 to receive the call set up information. The call flow manager 50 decides how a call is to be processed and instructs the switchboard and the

media service unit manager to configure the system and allocate resources of the system necessary to process the call. The scripts for carrying out the calls are stored in the SQL database 60.

The switchboard receives the call set up information packets which include, when available, information relating to the number dialed (DNI), caller ID (the caller's telephone number or ANI) as well as any other information as to the type of connection or the network the caller is calling from. The switchboard provides this information to the call flow manager which uses the information to select and process a script as well as make the information available to the scripts for processing the call. In addition, a caller already participating in an audio conference, such as a host, can initiate a script to place a call to a new participant of the audio conference. In this mode, the script gets the number from the host and instructs the switchboard to initiate a call to that number.

The call flow manager also determines how the system responds to various system events and uses the scripts to execute predefined responses to system events. The call flow manager receives the information from the switchboard relating to the system event and uses the information to select and execute the appropriate switch.

The media service unit (MSU) manager 30 manages the media service unit resources of units 35A-35N that provide auto conferencing services to the individual caller. The MSU manager communicated with the call flow manager and assigns the MSU system resources to each call that is processed. The MSU manager assigns the MSU system resources as a function of the MSU resource that is available. The web server 70 also provides a web-based interface to allow a participant to join an audio conference or allow the host of a conference to add or drop a participant from an audio conference. This interface is carried out by the server 70 which interfaces with the Internet 3 and provides information via the IP switch to the call flow manager 50 and the MSU manager 30.

The MSU manager also provides audio conferencing services such as enabling a plurality of audio conferences, real time conference recording and playback, and interactive voice response based services. In one embodiment, each media service unit 35A-35N is a 500 megahertz Intel Pentium III CPU with 512 megabytes of RAM. Each runs a Microsoft Windows NT server operating system, Microsoft Internet Information Server and Microsoft Transaction Server. In this configuration, the switchboard, call flow manager, and MSU manager, can be software modules compliant with the Microsoft Component Object Model and Distributed Component Object Model specifications. The scripts can be written in Microsoft Visual Basic.

The media service units receive call content information packets and either combine them with other packets to be sent to other participants in the audio conference, are sent to other participants of the audio conference or are discarded, depending upon the configuration of the audio conference. In one embodiment, where there are two or more participants in an audio

conference and participant 1 and participant 2 are the loudest speakers, the audio conferencing bridge can be configured to send the following signals to the following participants: participant 1, the signal received from participant 2; participant 2, the signal received from participant 1; and to participant 3 and all other participants, the combined signal from participant 1 and participant 2.

The call flow manager can be adapted to utilize a plurality of scripts that determine how incoming calls, based upon information received in the call signaling packets, can be processed. The choice of scripts can be determined by ANI/DNI, or by conference number or member number input by the caller. The call flow manager can process each incoming call by having the MSU manager assign the call to a media service unit which can play a prompt to the caller asking the caller to identify a conference to be connected to as well as a password or other authenticating identification. The caller can input information via a telephone key pad (DTMF) or by voice. Based upon the input information, the incoming call can be connected to the

media service unit responsible for that audio conference. Alternatively, the call flow manager can execute a script which notifies the host of a conference that a participant is waiting to be connected to the conference and await the host authorization to let the participant join the conference.

When a local telephone network notifies the bridge that a call has been placed to the audio conferencing switch, the switchboard notifies the call flow manager of the incoming call and provides any information that was included in the call signaling message received from the local exchange carrier. The call flow manager, using that information (such as the ANI/DNI), selects and executes the script to process the incoming call. During the course of executing the script, the call flow manager can instruct the MSU manager to assign the media service unit to service the incoming call. If the call flow manager is unable to determine which audio conference the incoming call is to be connected to, the system must query the caller to ascertain the identifier of the appropriate conference. Thus a first media service unit can be used to

query the caller to ascertain the appropriate conference and then the call may be assigned to a second media service unit that hosts the conference.

The switchboard and call flow manager also work together to provide services to participants of a conference, such as to allow a participant to exit from a conference or to switch conferences.

The system according to the present invention processes a call as follows. The incoming call from the local telephone network has the information therein converted into VOIP packets which are supplied via the gateway 5 to the bridge 100. The incoming packets are transmitted to the switchboard and when the switchboard answers the call, it notifies the call flow manager. The call flow manager notifies the MSU manager that a particular MSU service is needed and, in one embodiment, the MSU manager notifies the switchboard of the IP address of the MSU service that will service the incoming call. The switchboard then notifies the gateway of the port or the destination IP address from the media service unit that

will process the call. The IP switch directs the IP packets to the appropriate port which was assigned by the MSU manager to process the call. Once a call has been assigned to a media service unit, the media service unit processes all call content IP packets received from or sent to a caller. Thus the media service unit can add the caller to a conference, drop the caller from a conference or respond to an interactive voice response request. In one embodiment, the MSU manager is a DCOM object that spawns instances of the MSU 35A-35N to handle each call.

Alternatively, the IP switch can receive inputs from a web site maintained by servers 70 that allows management of an audio conference. The MSU manager receives instructions from the servers via the IP switch and acts on those instructions to add a caller to a conference, delete a caller from a conference, mute a caller or provide additional services. The inputs are generated by the host and optionally by the other participants by activating "buttons" on a monitor of computer 2A. An example of a screen with these controls is shown in Fig. 11.

The method can further comprise providing a web site relating to the audio conferencing services, allowing participants to access the web site and display images thereon as shown in Fig. 11 including a display of at least one of the participants on a display of at least one of the other participants of a conference call. The screen also provides conference call service options on a display of at least one participant and permits the at least one participant to select a conference call service option from the display. The options as shown include dialing out to additional participants, muting at least one of the participants and determining the number of participants in the conference call.

Each media service unit 35A-35N provides audio conferencing services to an assigned caller. Each caller is assigned an IP address of a media service unit that receives audio data IP packets and each media service unit is responsible for transmitting IP packets of call content information to the caller. In accordance with the invention, the media service units determine what audio information is output to the caller. In one embodiment,

where there are more than two participants to an audio conference, the conference can be configured to send the signals from one participant to the other and vice-versa and the combined signals from the two participants to a third participant. In an embodiment where there are only two participants, the conference is configured so that each hears what the other is saying.

Figures 2-10 show the flow of various methods of providing conferencing services in accordance with the present invention.

Figure 2 illustrates the method according to the present invention using the bridge shown in Figure 1 for the host calling in to set up in a conference.

In step 200, a caller dials a local number such as 1-617-yes-eYak, an 800 number or the like to reach the IP switch 10. Upon connection to the IP switch, there is an eYak chime in step 201 and a welcoming message in step 202. The call flow manager then initiates a script in step 203 where the caller is asked whether the caller wants to join a conference, host a conference, or sign up with the

system. If the caller wishes to sign up, the caller is sent to step 204 which transfers out to a caller sign in routine. If the caller presses no button or presses an incorrect button, an error message will be obtained in step 220. If the caller indicates a desire to join a conference, the caller is sent to a participant entry process in step 205. On the other hand if the caller chooses to host a conference, the caller will be asked for a conference identification in step 207. A host conference identification entry can also be achieved from another path at step 206 (step 407 in Fig. 4) according to the present invention. If the caller in step 207 enters an unrecognized identification, an error message will be given in step 208. If the caller does not have an identification but wishes to sign up for one, then the caller is sent to step 204. If the caller enters an identification, the caller is asked in step 209 to enter a pin number. If the pin number is incorrect, there will be an error message at step 210. If the pin number is correct, the system will next determine if the conference is underway in step 211. If the conference is underway, then the caller is sent to

the host reentry step 212. If the conference is not underway, a message is given to the caller in step 213 to set up the conference to see if a password is required. Step 213 can also be accessed via the host conference routine from step 214 (see step 407 in Fig. 4). If no password is required, then the caller hears the eYak chime in step 216. If the caller enters a password, a scripted response is given in step 215 and then the chime is received in step 216. After the chime, the caller is prompted with a starting message in step 217 and the conference is started with the host as the initial member in step 218.

Figure 3 illustrates the participant entry process for the method according to the present invention.

The participant entry process starts in step 300 whereupon the caller is prompted to enter the identification of the conference in step 301. If an improper identification is entered, an error message will be received in step 302. If a correct identification is received, then a determination must be made if the

conference has started already in step 303. If the conference has not started, the caller is informed of this fact in step 304 and a message is played or promotional material is heard by the caller for a given amount of time, preferably 10 minutes in step 305. If the conference does not start after ten minutes, the caller is informed that the conference host has not arrived and that the caller should try for the conference at another time in step 306, whereupon the caller is disconnected in step 307. If on the other hand the conference starts before the end of 10 minutes, a determination is made as to whether the conference requires a password in step 309. A host reentry as a participant can also occur at this point in the process from step 308. If no password is required, the eYak chime is received in step 311 and a scripted message is delivered to the caller indicating that the conference is being joined in step 312 and the participant is added to the conference in step 313 and the other participants are informed of this fact by the use of an entry tone. If the conference requires a password, the user is prompted to enter the conference password in step 310. If the password

is improper, an error message will be received in step 320. If the proper password is entered, the caller will be directed to step 311 as previously described.

If the conference has started, a determination is made in step 314 whether the conference is full. If the conference is full a scripted message is returned to the caller in step 315 informing the caller of this fact and the caller is disconnected in step 316. If the conference is not full, the next determination to be made is whether the conference is locked in step 317. If the conference is locked, a scripted message is delivered to the caller in step 318 informing the caller of this fact and the caller is disconnected in step 319. If the conference is not locked, the caller is directed to the step 309 as previously described.

Figure 4 illustrates the host reentry method starting at step 400. When the host seeks to reenter, the first determination to be made by the system is whether the existing conference still has a host connected and this is done in step 401. If there is no host connected, the

caller is prompted with the message in step 402 and the menu of items therein. If the caller wishes to rejoin the conference as the host, the caller is added to the conference as a host in step 403. If the caller wishes to end the existing conference and start a new one, the caller is directed to the menu items in step 404. If the caller inadvertently had asked to end the conference, the caller is directed back to the choices given in step 402. On the other hand, if the caller seeks to end the existing conference, this is carried out in step 405 and the caller is directed to the host conference setup in step 406.

If the caller sought to reenter the conference identification, then the host is directed to step 407 wherein the host identification entry routine is carried out.

Returning back to step 401, if the existing conference still has a host connected, the system then determines if the conference is full. If the conference is full, the caller is prompted with a message in step 412 informing the caller of that fact and giving the caller the

option to host a different conference. If the caller seeks to host a different conference, the caller is directed to the host conference identification entry step 407. If the conference is not full, then a determination is made in step 409 if the conference is locked. If the conference is locked, the script informs the caller of this fact in step 413 and gives the caller the option to host a different conference whereupon the caller will be directed to step 407. If the conference is not locked, the caller is given the menu choices in step 410 to either join the conference as a participant or host a new conference. If the caller decides to join the conference as a participant, the caller is added as a participant in step 411 and the current participants are informed by means of an entry tone. If the caller seeks to form a different conference, the caller is directed to step 407.

Figure 5 illustrates some of the teleconferencing services that may be performed in accordance with the method and system of the present invention. During the conference, a caller presses certain numbers in step 500 to access these functions. As noted, the caller can get to

any of the sub-functions by pressing the appropriate numbers directly. If the caller is a host, the caller is prompted with a menu in step 501 giving the caller a number of options. If the caller is a participant, the caller is prompted with a menu in step 502. The options for the participant in step 502 is to mute or unmute the line whereupon the lines is muted in step 505, to speak with a customer service representative, whereupon the user is transferred out in step 515, to hear the menu again, or to return to the conference whereupon the user is returned to the conference in step 503. If a button is hit which is incorrect, the error message will be received in step 504 and the caller will be returned to the conference. Similarly, if the caller does not react to any of the menu items, the caller will be returned to the conference in step 503.

The host has a number of other options available. One option is to dial out to a new participant which is carried out in step 506. The second option is to allow the participants to continue talking after the host disconnects and this is carried out in step 507. Another option is to

carry out call recording which is initiated in step 508. The host can also lock or unlock a conference and this is carried out in step 509. The host may initiate a roll call in step 510 or the host can mute a particular caller in step 511. The host can initiate a lecture mode, wherein all of the participants, with the exception of the host are muted and this is carried out in step 512. The host can record a conference greeting in step 513 or the host can change the pin number for the conference in step 514. Alternatively, the host can return to the conference at any time in step 503 or the host can go to a customer representative and transfer out in step 515.

Figure 6-10 show how the various functions shown in Figure 5 are carried out by the system.

In Figure 6, the dial out routine is shown starting at step 600 whereupon the specific number entries from the caller. The system determines in step 601 if the conference is full, and if not, the caller is prompted in step 602 to enter the area code and phone number of the person to be added and whether or not the host wants to

rejoin with the new participant or disconnect the new participant and rejoin without the new participant. The host dials the number and gives the information regarding the joiner which is received in step 603. If the host seeks to rejoin without the new participant, a chime is received in step 604, the user is prompted in step 605 that the caller is rejoining the conference and the new participant is abandoned and the host is returned to the conference in step 606 and the conference hears the appropriate entry tone. If on the other hand the host seeks to join with the new participant, an eYak chime is received in step 607, the announcement message is given in step 608 that they are rejoining and the host and new participant are added to the conference in step 609 and a suitable entry tone is sounded. If the conference is full as determined in step 601, the caller is prompted in step 610 of this fact and informed that before one can dial out to more participants, some of the participants must hang up. If the host is a paying subscriber as determined in step 611, the host is allowed to rejoin the conference as indicated by the prompt in step 613 and returned to the

conference in step 614. If the host is not a paying subscriber, the host is prompted in step 612 to increase the maximum size of the conference by signing up for a more expensive service.

The steps for carrying out conference continuation are shown starting at step 620 which leads to a determination by the system in step 621 if the conference is already set to continue after the host disconnects. If not, the host is prompted in step 622 with a message that the conference may now continue until the last participant hangs up even if the host is disconnected. The host is then returned to the conference in step 624. If the conference is already set up to continue after the host connects, the host is prompted in step 623 that the conference will be ended and that all participants will be disconnected when the host is disconnected. The host is then returned to the conference in step 624.

Figure 7 illustrates the steps for call recording, conference lock and roll call. If call recording is selected in step 700, a determination is made

by the system in step 701 if the conference is already being recorded. If not, the user is prompted in step 702 that conference recording will begin and the host is returned to the conference in step 703 with the message that conference recording has begun. If the conference is already being recorded, the host is prompted with the message that conference recording has been paused and to resume recording the user must initiate recording again. The host is then returned to the conference in step 705 and the message that the recording has stopped is played to the conference.

When the conference lock feature is selected in step 710, the system first determines in step 711 if the conference is already locked, and if not, the host is prompted with the message in step 712 that the conference is now locked and the host will be paged when new participants try to join the conference. The host is then returned to the conference in step 713. If the conference is already locked, the host will be prompted with the message in step 714 that the conference is now unlocked and to relock the host must go through the conference lock

routine again. The host is thereafter returned to the conference in step 713.

If the host selects the roll call function in step 720, the system first determines the number of people in the conference in step 721. If only one, the host is prompted with the message in step 722 that the host is the only person in the conference and the host is then returned to the conference in step 725. If there is more than one person in the conference, the host is prompted with the message in step 723 that there are N people in the conference, where N is the number of people in the conference. The host is then returned to the conference in step 725. In a preferred embodiment of the present invention, name recording is made available so that not only will the number of people in the conference be told to the host, but the roll call will include the names thereof.

Figure 8 shows the routines for mute calling and lecture mode. When the host selects to mute a caller in step 800, the system determined in step 801 is the caller is already muted. If not, the host is prompted with the

message in step 802 that the line is now muted and to unmute it the host must go through the process again. If the line is already muted, the host is prompted in step 803 that the line is now unmuted and to remute the line the user must go through the same process again. The caller is then returned to the conference in step 804. It will be apparent that this mute caller routine can also be used by the caller who is a participant.

When lecture mode is selected at step 810, the system first determines in step 811 if the participants are already muted. If not, the host is prompted with the message that all participants are now muted in step 812 and to unmute the host must go through the routine again. The host is then returned to the conference in step 814. If the participants are already muted, the host is prompted in step 813 that the participant lines have been unmuted and to re- mute them the host must be go through the process again. The host is then returned to the conference in step 814.

Figure 9 shows the steps for recording a greeting when such is selected at step 900. The host is prompted in step 901 to record a personal conference greeting and the caller can select the standard greeting or record a personal greeting. If the caller elects to use a standard greeting, the caller is prompt in step 907 that a standard greeting will be used and the host is returned to the conference in step 908. If the host elects to record a personal greeting, the host is prompted in step 902 to record the greeting at the tone. If the message exceeds 3 minutes, the host will be prompted in step 904 that the greeting went beyond the time limit. If the host does not enter any greeting, an error message at step 903 will be given to the host. If on the other hand the host records the message properly, the host is then prompted in step 905 with a menu to keep the greeting, erase the greeting, review the greeting or switch to a standard greeting and finally to return to the conference with the greeting unchanged. If no choice is made or an improper choice is made an error message is given in step 906. If the host seeks to use a standard greeting, the host is prompted in

step 907 that the standard greeting will be received as hereinbefore described. If the host seeks to keep the greeting, the host will be prompted in step 909 that the greeting will be kept and the host is returned to the conference. If the host decides to use the old greeting, the host will be prompted in step 910 that the old greeting will be kept and the host is returned to the conference. If the host wants to review the greeting, the host plays the greeting for review in step 911 and returns to the menu again as in step 905.

Figure 10 illustrates the steps for carrying out a change in the pin number for the conference. If change pin number is selected at step 1000, the system prompts the host in step 1001 to enter a new pin followed by the pound sign. If the host changes his or her mind, the host is returned to the conference in step 1002. If an improper pin number is entered, the host will receive an error message in step 1006. If a new pin number is entered, the host will be prompted to reenter the pin number in step 1003 and if it is incorrectly done the host will receive an error message in step 1004. If the second entry of the pin

number has been made, the host will be prompted in step 1005 that the new pin number has been recorded and the host will be returned to the conference in step 1007.

In accordance with a further aspect of the present invention, the operator of the conference bridge can bill for the audio conferencing services in a number of ways.

For example, the operator of the bridge can bill the host for each participant minute. That is the standard way that audio conferencing has always been billed. Alternatively, the operator of the bridge can bill each participant per minute of connection to a conference. Additionally, the conference bridge can play advertisements at selective times during the conference call or show advertisements on the computer screen of the user, as is shown in Figure 11.

In an alternative embodiment of the present invention, the operator of the bridge can make money as a result of the way in which the telephone industry is regulated. If a person makes a long distance call, the long distance company has to pay the local telephone

company that terminates the call. The bridge according to the present invention will have a local telephone company associated therewith. Thus each person calling the conference number will not have to pay for the conference, since the bridge will be receiving the payment from the long distance companies for terminating each of the long distance calls.

It is understood that the embodiments described hereinabove are merely illustrative and are not intended to limit the scope of the invention. It is realized that various changes, alterations, rearrangements and modifications can be made by those of skill in the art without substantially departing from the spirit and the scope of the present invention.

BNSDOCID: <WO 0172022A1 >

What is claimed is:

1. A method of providing audio conferencing services, comprising the steps of:

a. receiving Internet protocol packets containing call set-up information and Internet protocol packets containing call content information for at least one first participant at a first telephone number; and

b. setting up a conference call including the at least one first participant and at least one additional participant at a second telephone number based solely on the Internet protocol packets.

2. The method according to claim 1, wherein the step of setting up the conference call comprises channeling all of the Internet protocol packets through an Internet protocol switch.

3. The method according to claim 2, wherein the step of setting up the conference call further comprises receiving the call set-up information Internet protocol packets in a call flow manager to lead the first participant through a set-up procedure and receiving the content information Internet protocol packets in a media

service unit manager to provide a flow of content information between the first participant and additional participants.

4. The method according to claim 3, wherein the step of leading the first participant comprises running at least one of a plurality of scripts in the call flow manager.

5. The method according to claim 4, wherein the step of providing a flow of content information comprises controlling a plurality of media service units each handling a plurality of lines corresponding to the first and second telephone numbers.

6. The method according to claim 1, wherein the first participant is a host of the conference call and wherein the host is provided with conference call service options for a conference call.

7. The method according to claim 6, wherein the host is asked to enter a telephone number of an additional participant and the telephone number of the additional participant is called to add same to the conference call.

8. The method according to claim 6, wherein the host is permitted to mute and unmute at least one of the participants.

9. The method according to claim 6, wherein the host is permitted to mute and unmute all of the additional participants.

10. The method according to claim 6, wherein the host is permitted to query as to the number of participants in the conference call and the number is returned to the host.

11. The method according to claim 1, further comprising providing a web site relating to the audio conferencing services, allowing participants to access the web site and display images thereon and providing images from a display of at least one of the participants on a display of at least one of the other participants of a conference call.

12. The method according to claim 11, further comprising providing conference call service options on a display of at least one participant and permitting the at least one participant to select a conference call service option from the display.

13. The method according to claim 12, wherein the options comprise at least one of dialing out to additional participants, muting at least one of the participants and determining the number of participants in the conference call.

14. A system for providing audio conferencing services, comprising: a conference bridge having an Internet protocol switch receptive of Internet protocol packets containing call set-up information and Internet protocol packets containing call content information for at least one first participant at a first telephone number and at least one processor for setting up a conference call including the at least one first participant and at least one additional participant at a second telephone number based solely on the Internet protocol packets.

15. The system according to claim 14, wherein the conference bridge channels all of the Internet protocol packets through the Internet protocol switch.

16. The system according to claim 15, wherein the at least one processor comprises a call flow manager for receiving the call set-up information Internet protocol

packets to lead the first participant through a set-up procedure and a media service unit manager for receiving the content information Internet protocol packets to provide a flow of content information between the first participant and additional participants.

17. The system according to claim 16, wherein the call flow manager runs at least one of a plurality of scripts to lead the first participant.

18. The system according to claim 17, further a plurality of media service units each handling a plurality of lines corresponding to the first and second telephone numbers to provide the flow of content information.

19. The system according to claim 14, wherein the first participant is a host of the conference call and wherein the conference bridge provides the host with conference call service options for a conference call.

20. The system according to claim 19, wherein the conference bridge asks the host to enter a telephone number of an additional participant and the bridge dials out to the telephone number of the additional participant to add same to the conference call.

21. The system according to claim 19, wherein the conference bridge permits the host to mute and unmute at least one of the participants.

22. The system according to claim 19, wherein the conference bridge permits the host to mute and unmute all of the additional participants.

23. The system according to claim 19, wherein the conference bridge permits the host to query as to the number of participants in the conference call and returns the number to the host.

24. The system according to claim 14, wherein the conference bridge further comprises servers providing a web site relating to the audio conferencing services and accessible by participants to conference calls to display images on displays thereof and wherein the conference bridge provides images from a display of at least one of the participants on a display of at least one of the other participants of the conference call.

25. The system according to claim 24, wherein the conference bridge provides conference call service options on a display of at least one participant and permits the at

RNSDOCID: <WO 0172022A1 | >

least one participant to select a conference call service option from the display.

26. The system according to claim 25, wherein the options comprise at least one of dialing out to additional participants, muting at least one of the participants and determining the number of participants in the conference call.

27. A conference bridge comprising: an Internet protocol based telephone switch; a switchboard for receiving information from the switch to receive calls and dial out; a call flow manager receptive of call set up Internet protocol packets from the switch for leading a caller through a conference call set up procedure; and a media service unit manager receptive of call content Internet protocol packets from the switch for providing call content information flow between conference call participants.

28. The conference bridge according to claim 27, wherein the media service unit manager manages a plurality of media service units which are scalable in number to handle an increased number of lines.

3NSDOCID: <WO___0172022A1_I_>

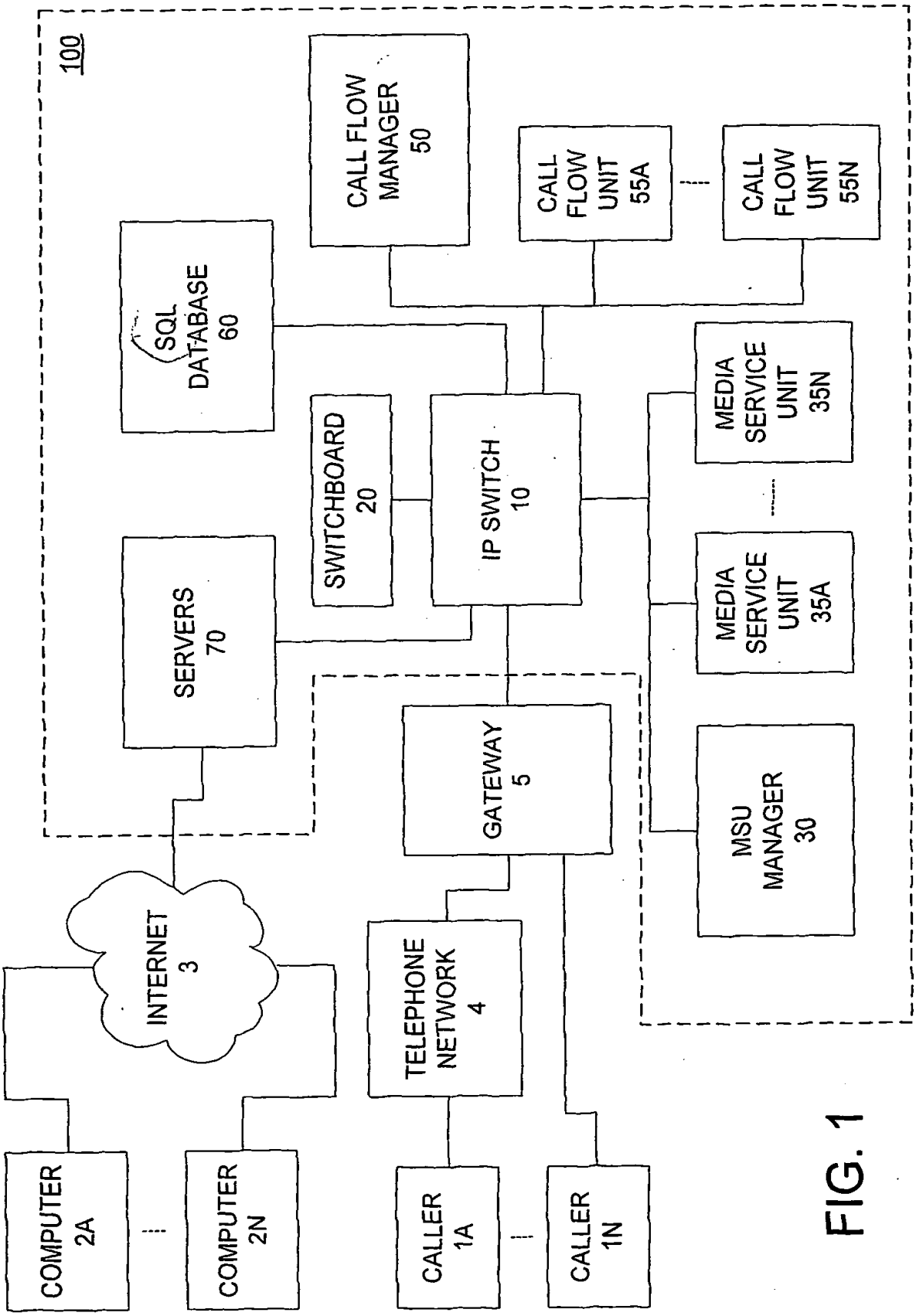

100
CALL FLOW MANAGER 50
CALL FLOW UNIT 55A
CALL FLOW UNIT 55N
SQL DATABASE 60
SWITCHBOARD 20
IP SWITCH 10
MEDIA SERVICE UNIT 35N
MEDIA SERVICE UNIT 35A
SERVERS 70
GATEWAY 5
MSU MANAGER 30
INTERNET 3
TELEPHONE NETWORK 4
COMPUTER 2A
COMPUTER 2N
CALLER 1A
CALLER 1N

FIG. 1

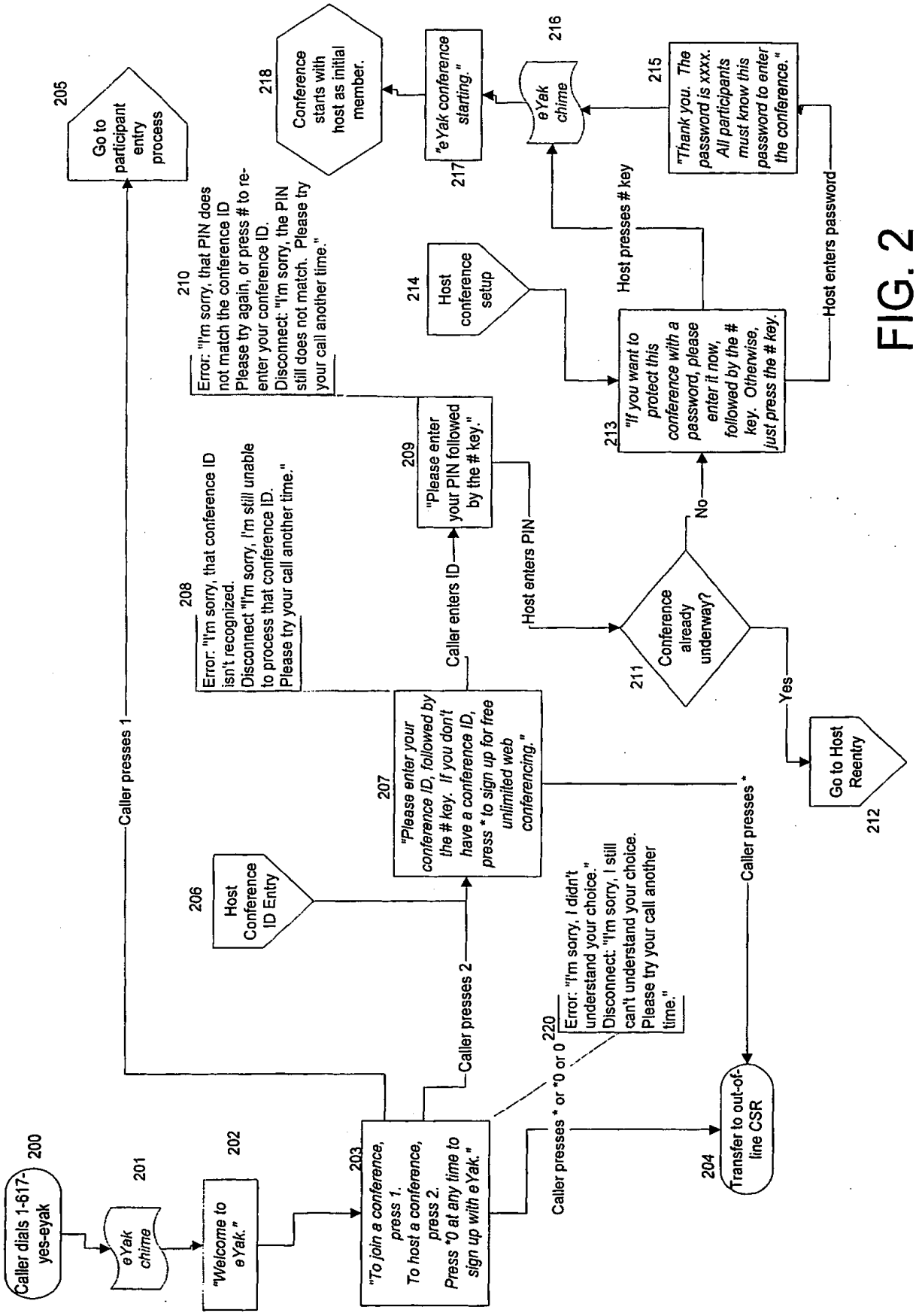

Caller dials 1-617-yes-eyak
200
eYak chime
201
"Welcome to eYak."
202
"To join a conference, press 1. To host a conference, press 2. Press *0 at any time to sign up with eYak."
203
Caller presses * or *0 or 0
Transfer to out-of-line CSR
204
Go to participant entry process
205
Caller presses 1
Host Conference ID Entry
206
Caller presses 2
"Please enter your conference ID, followed by the # key. If you don't have a conference ID, press * to sign up for free unlimited web conferencing."
207
Caller presses *
Error: "I'm sorry, I didn't understand your choice." Disconnect: "I'm sorry, I still can't understand your choice. Please try your call another time."
220
Error: "I'm sorry, that conference ID isn't recognized. Disconnect "I'm sorry, I'm still unable to process that conference ID. Please try your call another time."
208
Caller enters ID
"Please enter your PIN followed by the # key."
209
Error: "I'm sorry, that PIN does not match the conference ID Please try again, or press # to re-enter your conference ID. Disconnect: "I'm sorry, the PIN still does not match. Please try your call another time."
210
Host enters PIN
Conference already underway?
211
No
Yes
Go to Host Reentry
212
"If you want to protect this conference with a password, please enter it now, followed by the # key. Otherwise, just press the # key."
213
Host conference setup
214
Host presses # key
Host enters password
"eYak conference starting."
217
eYak chime
216
"Thank you. The password is xxxx. All participants must know this password to enter the conference."
215
Conference starts with host as initial member.
218

FIG. 2

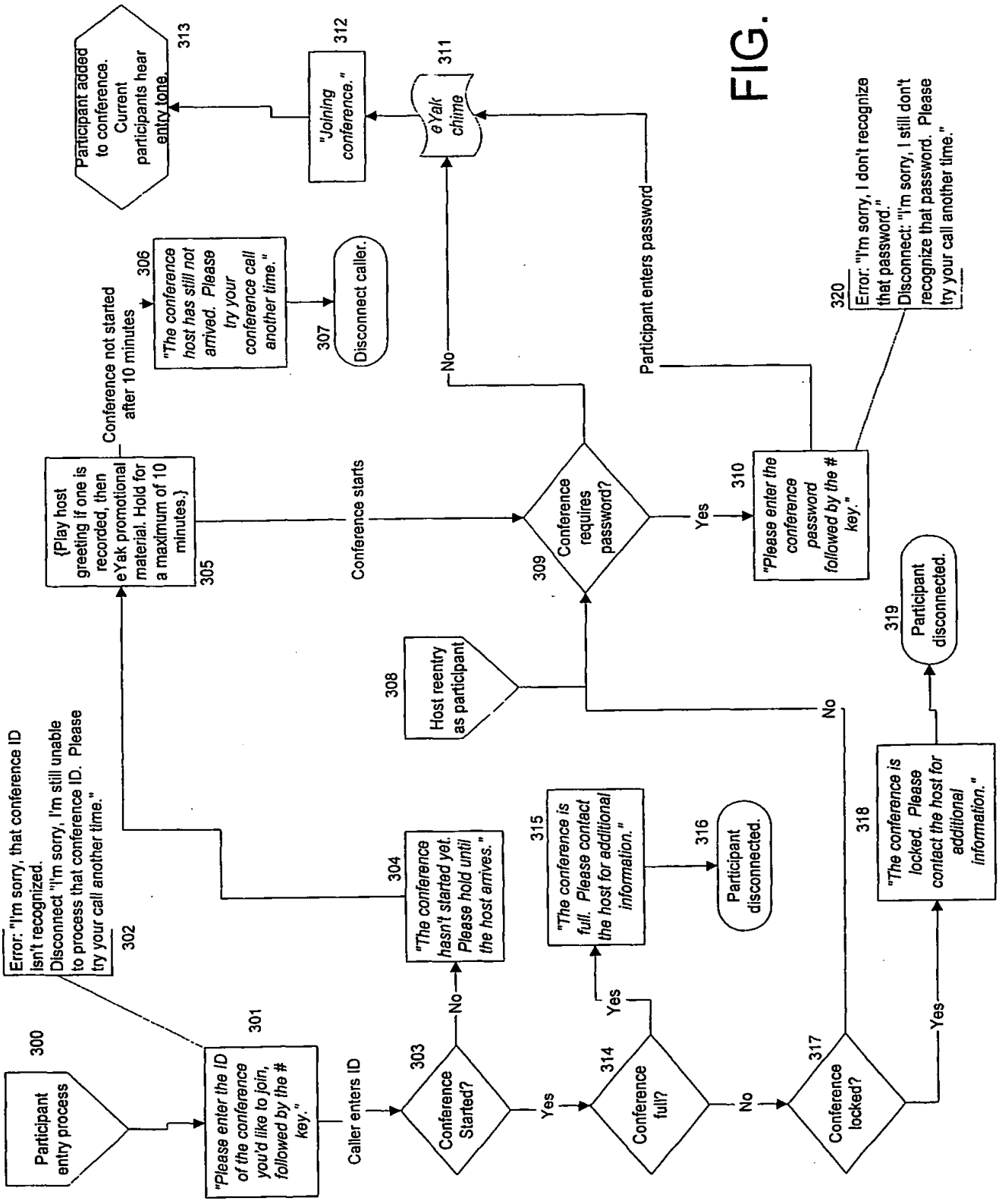
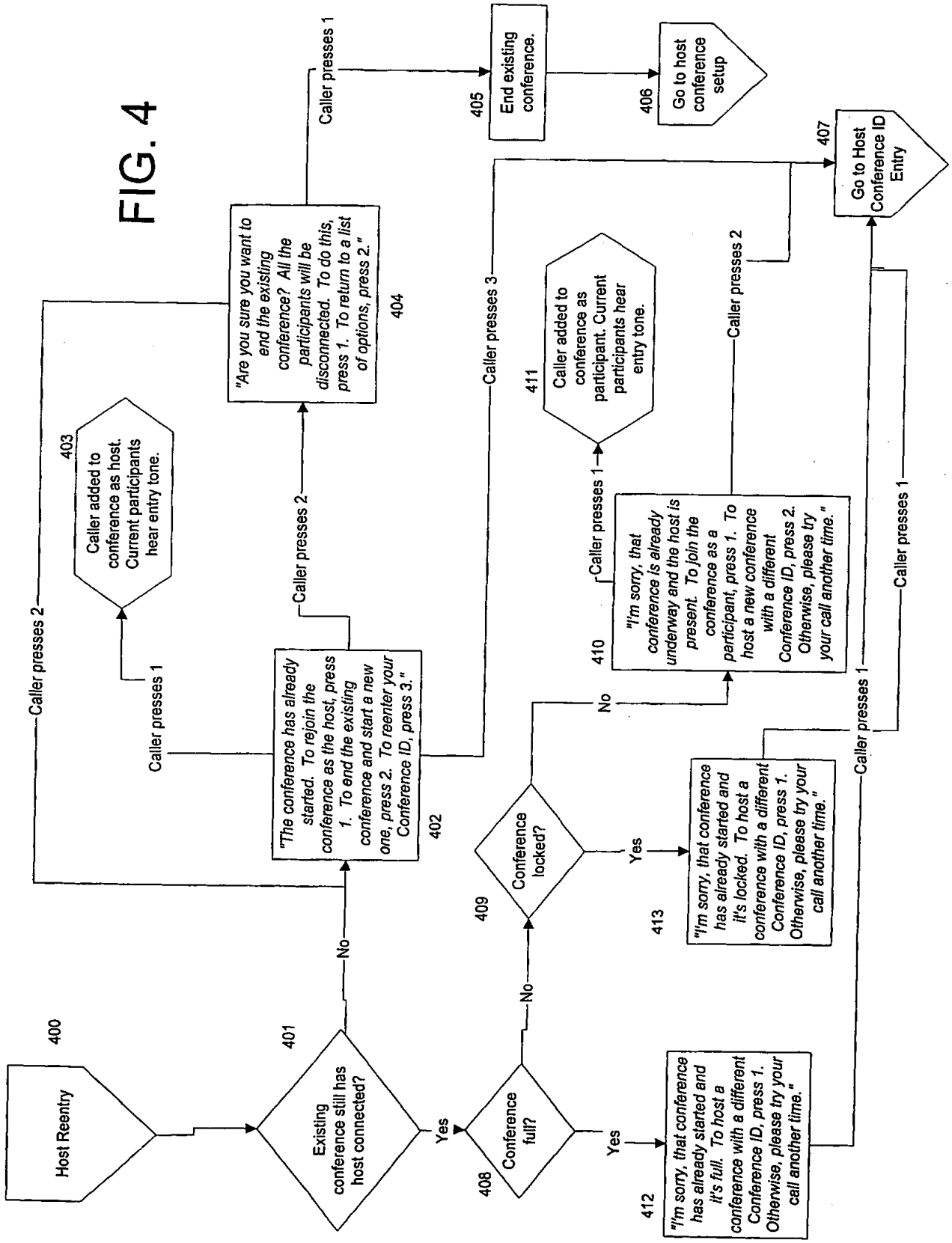

Participant entry process
300
"Please enter the ID of the conference you'd like to join, followed by the # key."
301
Caller enters ID
Conference Started?
303
No
Yes
Conference full?
314
Yes
No
Conference locked?
317
Yes
No
Error: "I'm sorry, that conference ID isn't recognized. Disconnect "I'm sorry, I'm still unable to process that conference ID. Please try your call another time."
302
"The conference hasn't started yet. Please hold until the host arrives."
304
"The conference is full. Please contact the host for additional information."
315
Participant disconnected.
316
"The conference is locked. Please contact the host for additional information."
318
Participant disconnected.
319
Host reentry as participant
308
{Play host greeting if one is recorded, then eYak promotional material. Hold for a maximum of 10 minutes.}
305
Conference not started after 10 minutes
"The conference host has still not arrived. Please try your conference call another time."
306
Disconnect caller.
307
Conference starts
Conference requires password?
309
Yes
"Please enter the conference password followed by the # key."
310
Participant enters password
Error: "I'm sorry, I don't recognize that password." Disconnect: "I'm sorry, I still don't recognize that password. Please try your call another time."
320
No
eYak chime
311
"Joining conference."
312
Participant added to conference. Current participants hear entry tone
313

FIG. 3

FIG. 4



BNSDOCID: <WO 0172022A1 >

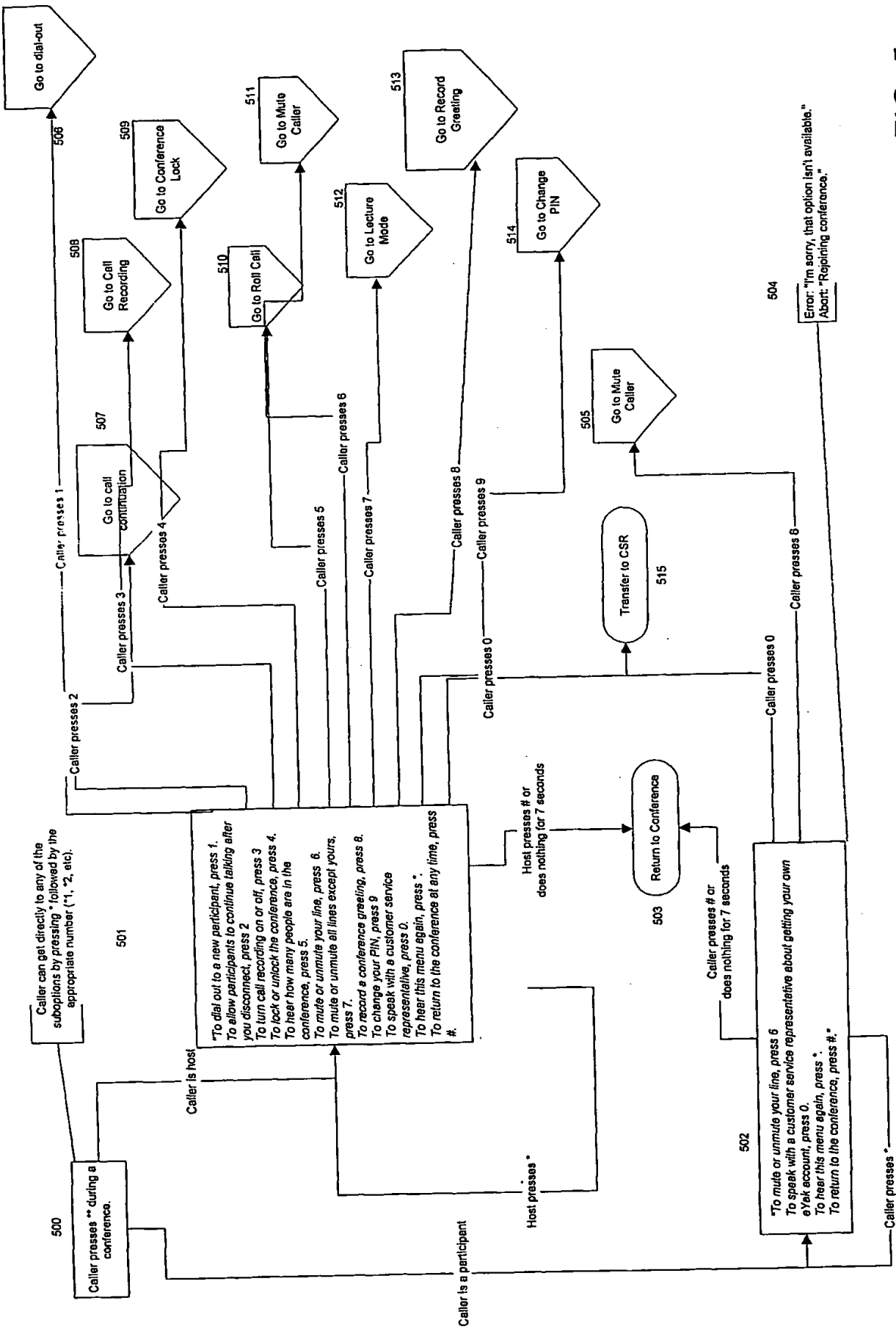

Caller can get directly to any of the suboptions by pressing * followed by the appropriate number (*1, *2, etc).
501
"To dial out to a new participant, press 1.
To allow participants to continue talking after you disconnect, press 2
To turn call recording on or off, press 3
To lock or unlock the conference, press 4.
To hear how many people are in the conference, press 5.
To mute or unmute your line, press 6.
To mute or unmute all lines except yours, press 7.
To record a conference greeting, press 8.
To change your PIN, press 9
To speak with a customer service representative, press 0.
To hear this menu again, press *.
To return to the conference at any time, press #.
Caller presses ** during a conference.
500
Caller is host
Caller is a participant
Host presses *
Host presses # or does nothing for 7 seconds
Return to Conference
503
Caller presses # or does nothing for 7 seconds
"To mute or unmute your line, press 6
To speak with a customer service representative about getting your own eYak account, press 0.
To hear this menu again, press *.
To return to the conference, press #."
502
Caller presses *
Error: "I'm sorry, that option isn't available."
Abort: "Rejoining conference."
504
Go to Mute Caller
505
Transfer to CSR
515
Go to call continuation
507
Go to Call Recording
508
Go to Conference Lock
509
Go to dial-out
506
Go to Roll Call
510
Go to Mute Caller
511
Go to Lecture Mode
512
Go to Record Greeting
513
Go to Change PIN
514
Caller presses 1
Caller presses 2
Caller presses 3
Caller presses 4
Caller presses 5
Caller presses 6
Caller presses 7
Caller presses 8
Caller presses 9
Caller presses 0
Caller presses 0
Caller presses 8

FIG. 5

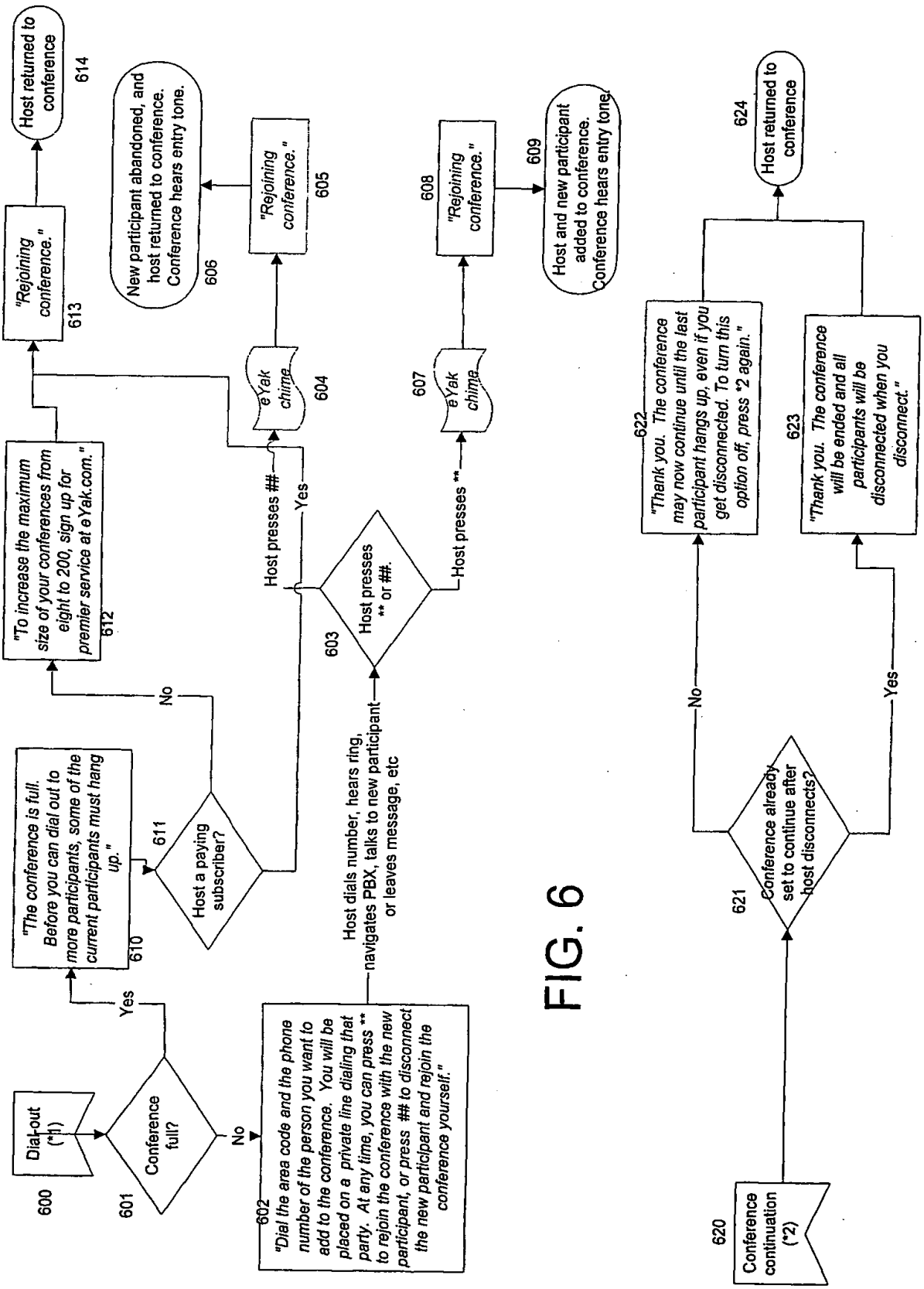

600
Dial-out (*1)
601
Conference full?
Yes
No
602
"Dial the area code and the phone number of the person you want to add to the conference. You will be placed on a private line dialing that party. At any time, you can press ** to rejoin the conference with the new participant, or press ## to disconnect the new participant and rejoin the conference yourself."
Host dials number, hears ring, navigates PBX, talks to new participant or leaves message, etc
603
Host presses ** or ##.
Host presses ##
Yes
Host presses **
604
eYak chime
605
"Rejoining conference."
606
New participant abandoned, and host returned to conference. Conference hears entry tone.
607
eYak chime
608
"Rejoining conference."
609
Host and new participant added to conference. Conference hears entry tone.
610
"The conference is full. Before you can dial out to more participants, some of the current participants must hang up."
611
Host a paying subscriber?
No
612
"To increase the maximum size of your conferences from eight to 200, sign up for premier service at eYak.com."
613
"Rejoining conference."
614
Host returned to conference
620
Conference continuation (*2)
621
Conference already set to continue after host disconnects?
No
622
"Thank you. The conference may now continue until the last participant hangs up, even if you get disconnected. To turn this option off, press *2 again."
623
"Thank you. The conference will be ended and all participants will be disconnected when you disconnect."
Yes
624
Host returned to conference

FIG. 6

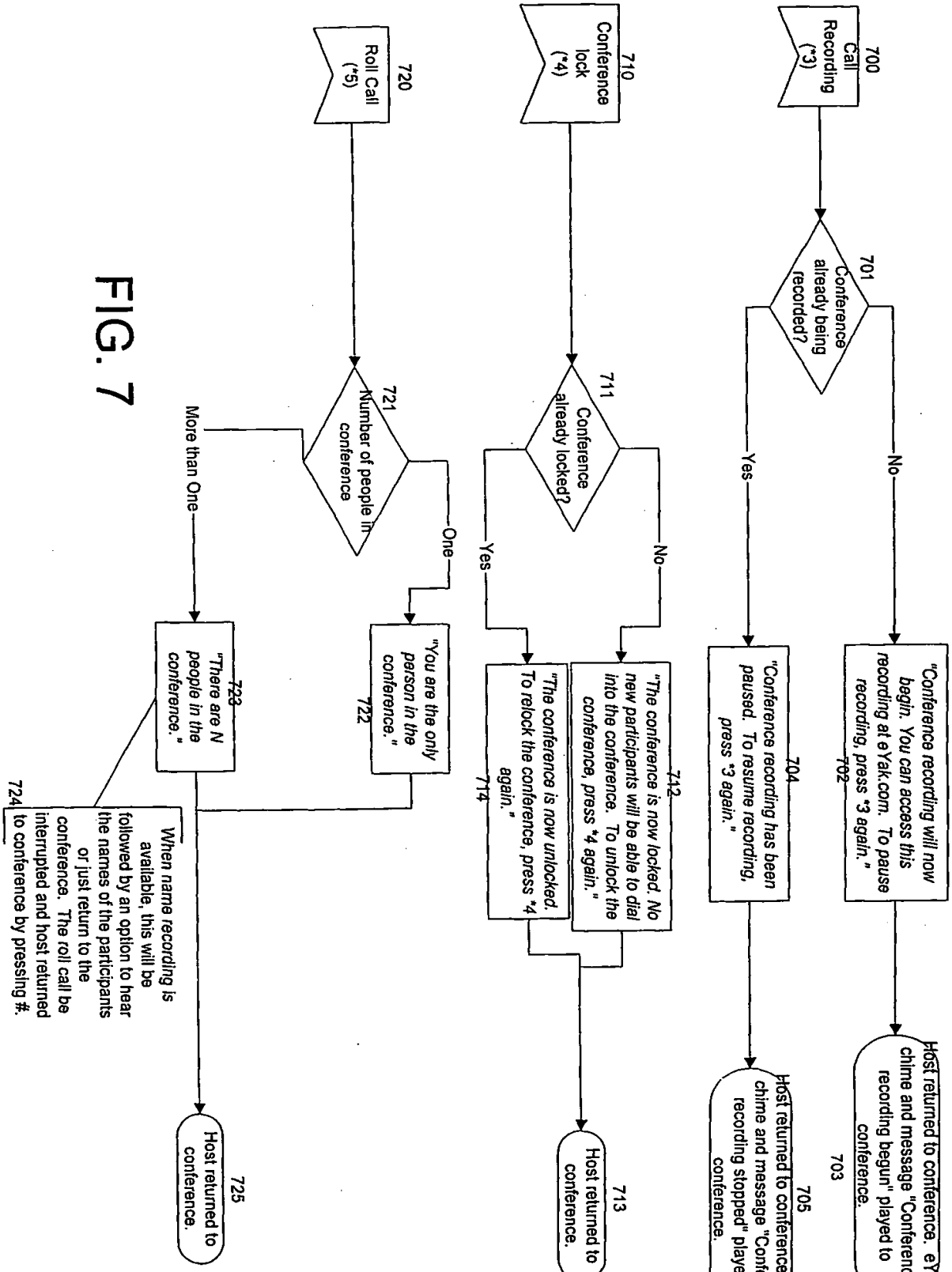

700
Call Recording (*3)
701
Conference already being recorded?
No
Yes
"Conference recording will now begin. You can access this recording at eYak.com. To pause recording, press *3 again."
702
Host returned to conference. eYak chime and message "Conference recording begun" played to conference.
703
"Conference recording has been paused. To resume recording, press *3 again."
704
Host returned to conference. eYak chime and message "Conference recording stopped" played to conference.
705
710
Conference lock (*4)
711
Conference already locked?
No
Yes
"The conference is now locked. No new participants will be able to dial into the conference. To unlock the conference, press *4 again."
712
Host returned to conference.
713
"The conference is now unlocked. To relock the conference, press *4 again."
714
720
Roll Call (*5)
721
Number of people in conference
One
More than One
"You are the only person in the conference."
722
"There are N people in the conference."
723
When name recording is available, this will be followed by an option to hear the names of the participants or just return to the conference. The roll call be interrupted and host returned to conference by pressing #.
724
Host returned to conference.
725

FIG. 7

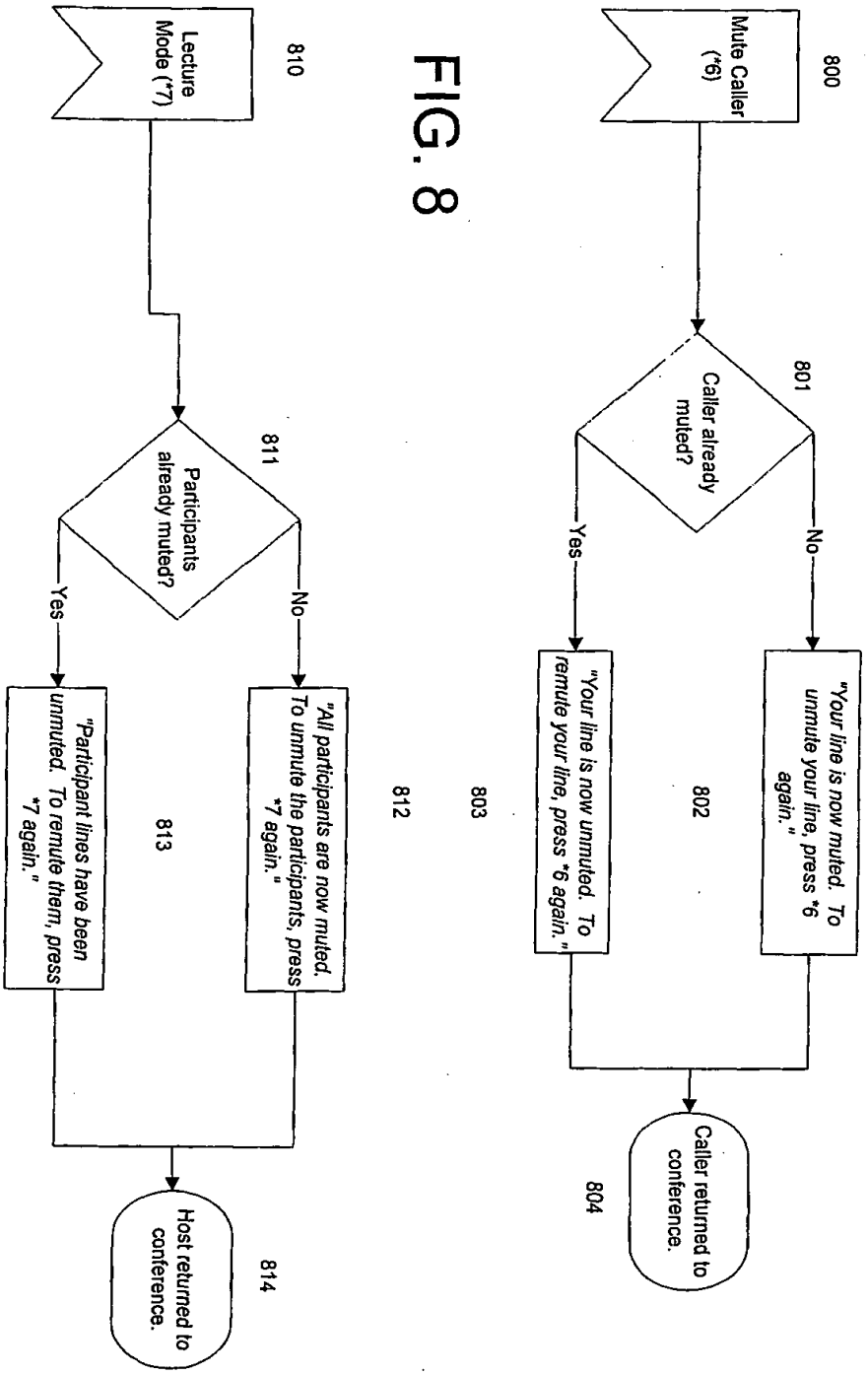

Mute Caller (*6)
800
Caller already muted?
801
Yes
No
"Your line is now unmuted. To remute your line, press *6 again."
803
"Your line is now muted. To unmute your line, press *6 again."
802
Caller returned to conference.
804
Lecture Mode (*7)
810
Participants already muted?
811
Yes
No
"Participant lines have been unmuted. To remute them, press *7 again."
813
"All participants are now muted. To unmute the participants, press *7 again."
812
Host returned to conference.
814

FIG. 8

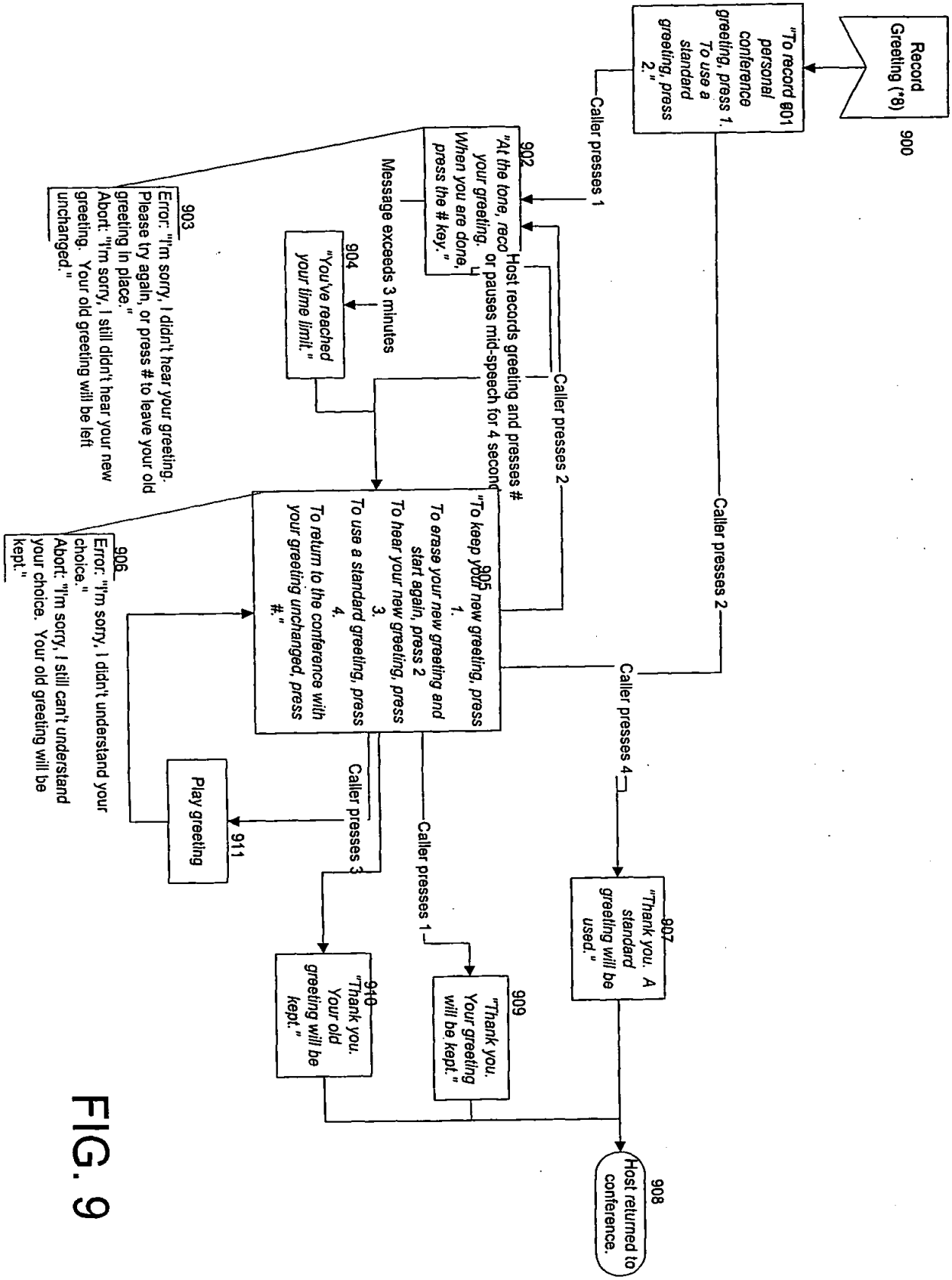

Record Greeting (*8)
900
"To record a personal conference greeting, press 1. To use a standard greeting, press 2."
Caller presses 1
Caller presses 2
902
"At the tone, record your greeting. When you are done, press the # key."
Host records greeting and presses # or pauses mid-speech for 4 seconds
Caller presses 2
Message exceeds 3 minutes
904
"You've reached your time limit."
903
Error: "I'm sorry, I didn't hear your greeting. Please try again, or press # to leave your old greeting in place."
Abort: "I'm sorry, I still didn't hear your new greeting. Your old greeting will be left unchanged."
905
"To keep your new greeting, press 1. To erase your new greeting and start again, press 2 To hear your new greeting, press 3. To use a standard greeting, press 4. To return to the conference with your greeting unchanged, press #."
906
Error: "I'm sorry, I didn't understand your choice."
Abort: "I'm sorry, I still can't understand your choice. Your old greeting will be kept."
Play greeting
911
Caller presses 3
Caller presses 1
Caller presses 4
907
"Thank you. A standard greeting will be used."
909
"Thank you. Your greeting will be kept."
910
"Thank you. Your old greeting will be kept."
908
Host returned to conference.

FIG. 9

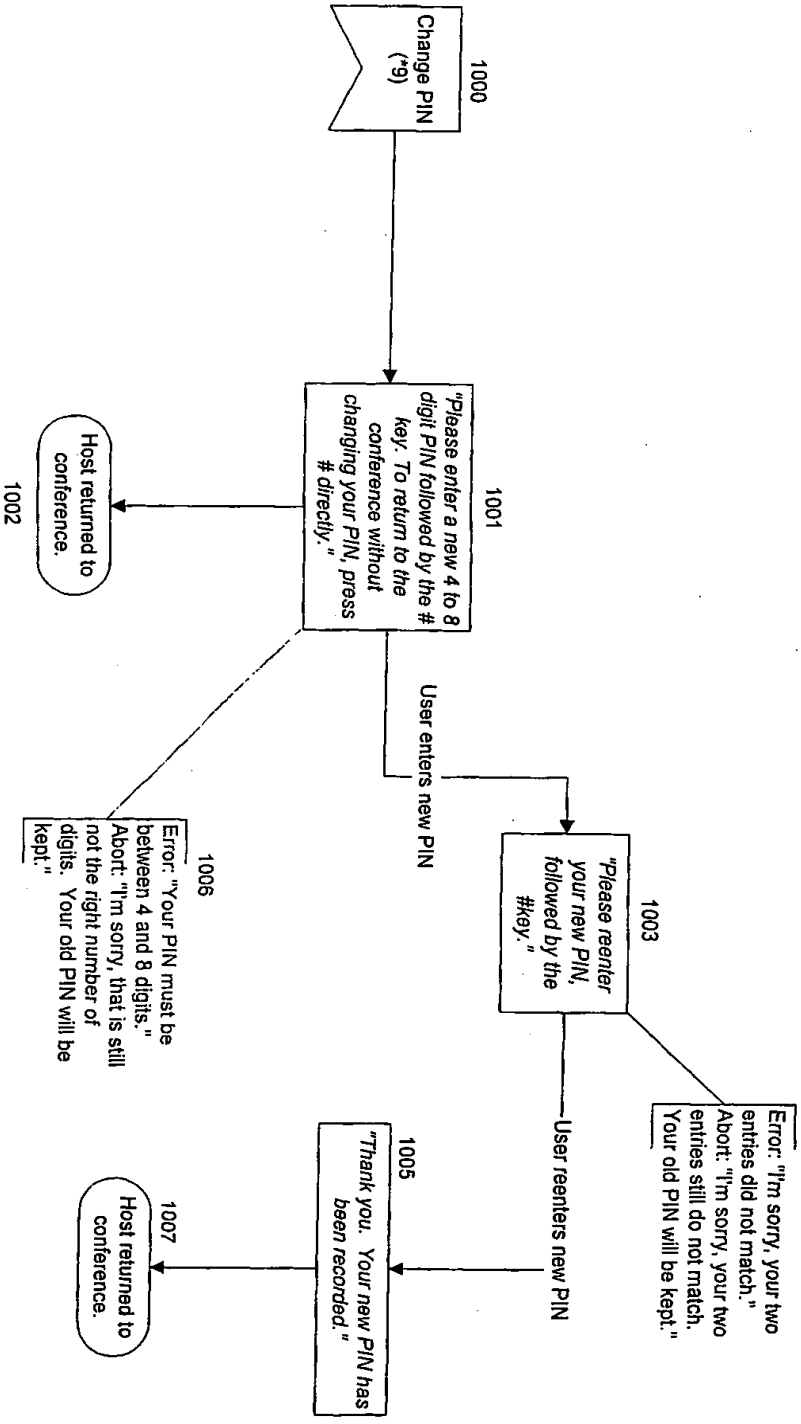

Change PIN (*9)
1000
"Please enter a new 4 to 8 digit PIN followed by the # key. To return to the conference without changing your PIN, press # directly."
1001
Host returned to conference.
1002
User enters new PIN
"Please reenter your new PIN, followed by the #key."
1003
Error: "I'm sorry, your two entries did not match." Abort: "I'm sorry, your two entries still do not match. Your old PIN will be kept."
1004
User reenters new PIN
"Thank you. Your new PIN has been recorded."
1005
Host returned to conference.
1007
Error: "Your PIN must be between 4 and 8 digits." Abort: "I'm sorry, that is still not the right number of digits. Your old PIN will be kept."
1006

FIG. 10

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US01/08082

A. CLASSIFICATION OF SUBJECT MATTER
IPC(7) :H04M 3/42
US CL :379/202
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED
Minimum documentation searched (classification system followed by classification symbols)
U.S. : 379/202-205; 370/261; 709/204

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)
EAST

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US 5,884,032 A (BATEMAN et al) 16 March 1999, see entire patent.	1-28
A,P	US 6,167,432 A (JIANG) 26 December 2000, see entire patent.	1-28
A,P	US 6,173,314 B1 (KURASHIMA et al) 09 January 2001, see entire patent.	1-28

☐ Further documents are listed in the continuation of Box C. ☐ See patent family annex.

*	Special categories of cited documents:	"T"	later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
"A"	document defining the general state of the art which is not considered to be of particular relevance	"X"	document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
"E"	earlier document published on or after the international filing date	"Y"	document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art
"L"	document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)	"&"	document member of the same patent family
"O"	document referring to an oral disclosure, use, exhibition or other means		
"P"	document published prior to the international filing date but later than the priority date claimed		

Date of the actual completion of the international search
02 MAY 2001

Date of mailing of the international search report
07 JUN 2001

Name and mailing address of the ISA/US
Commissioner of Patents and Trademarks
Box PCT
Washington, D.C. 20231
Facsimile No. (703) 305-3230

Authorized officer
AHMAD MATAR Eugene Zogan
Telephone No. (703) 305-4731

Form PCT/ISA/210 (second sheet) (July 1998) ★

This Page Blank (uspto)